

FIG. 1

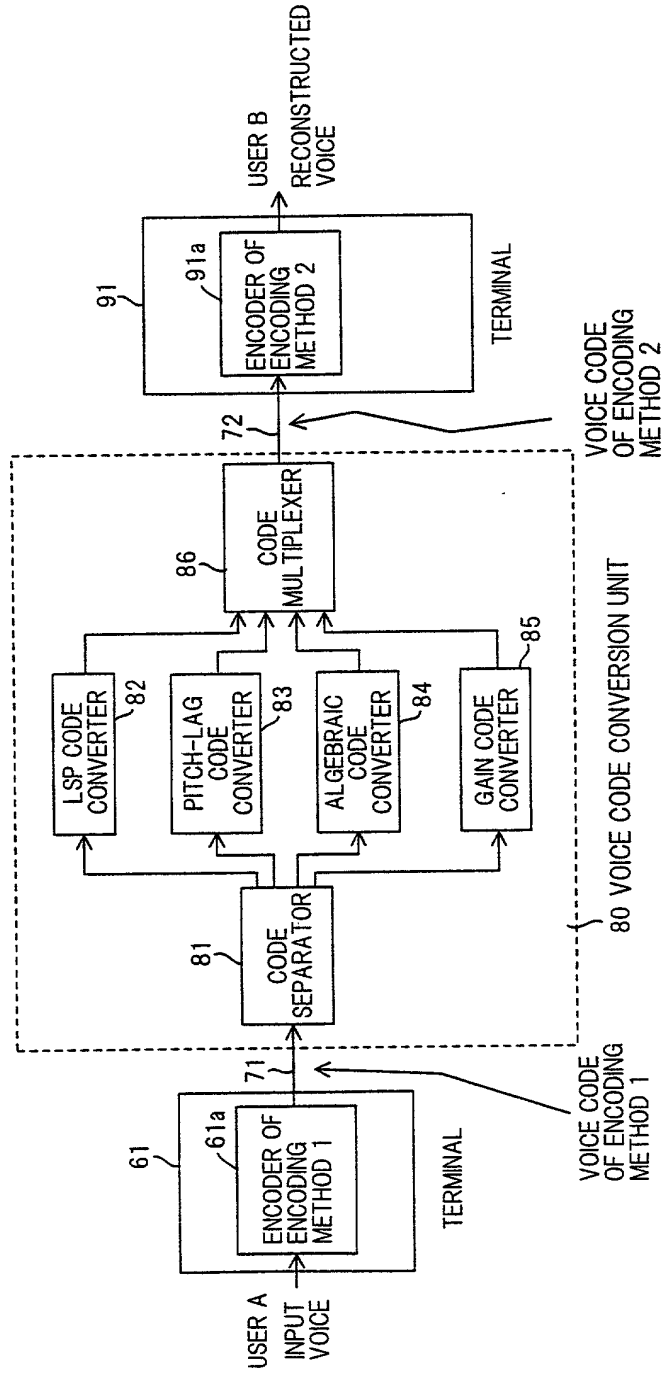


FIG. 2

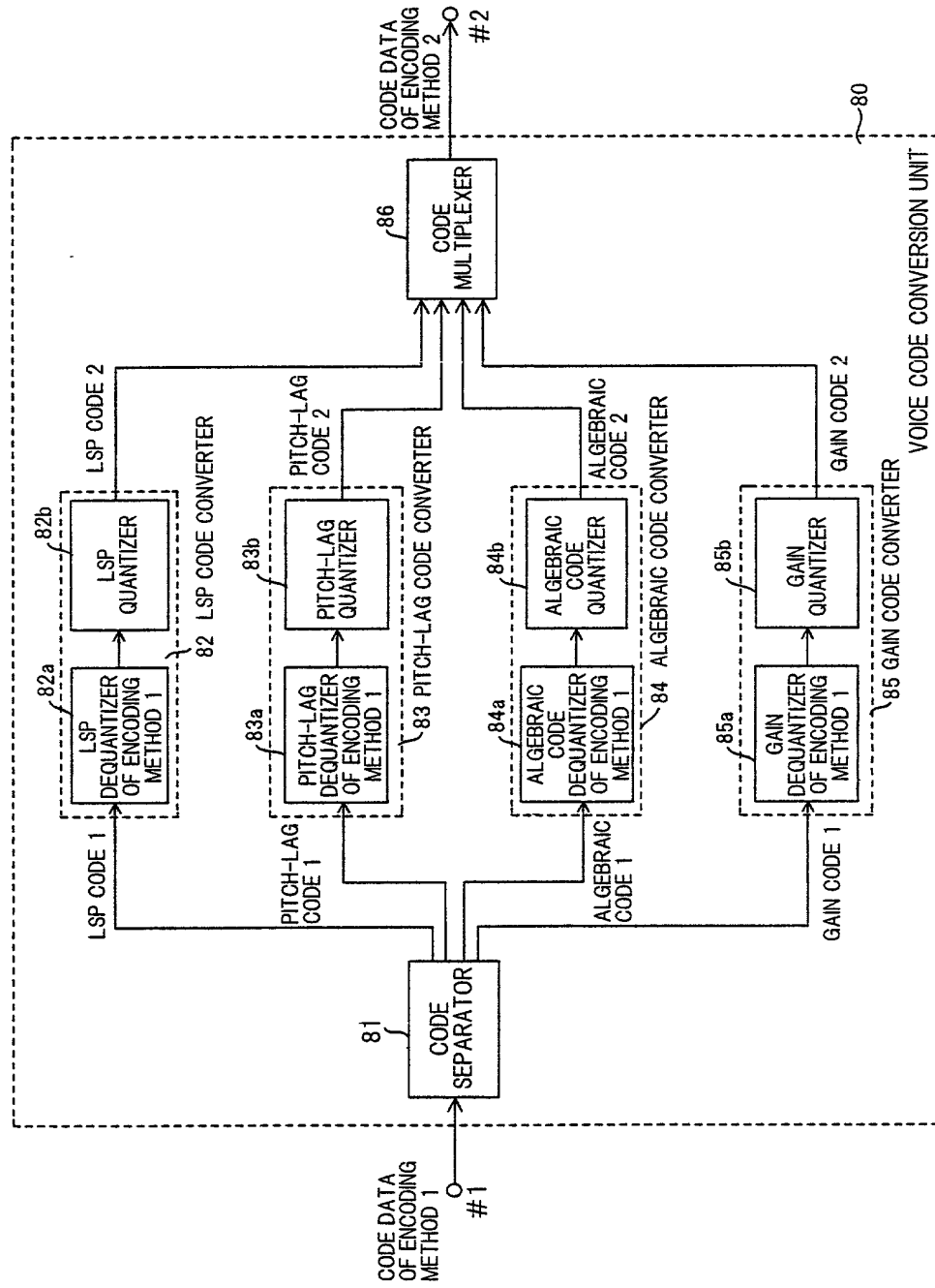


FIG. 3

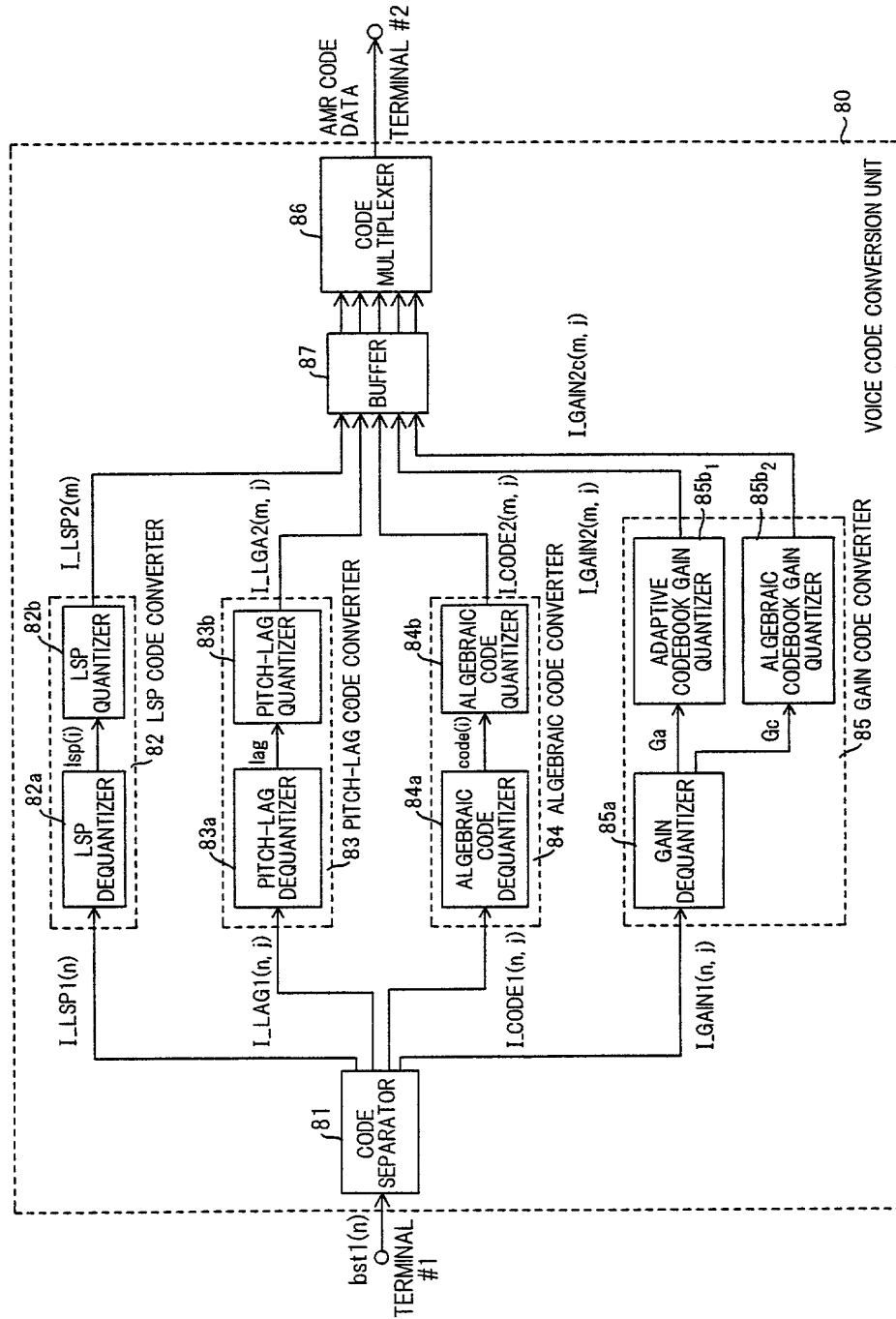
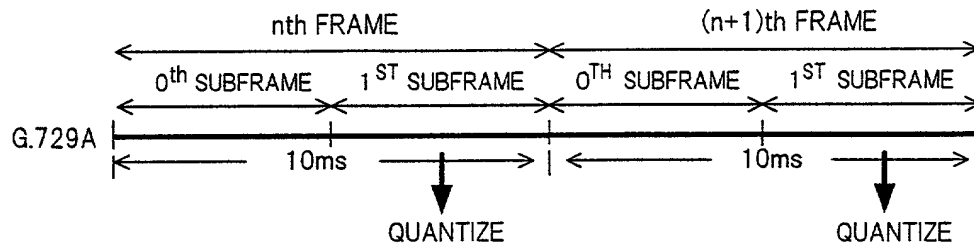
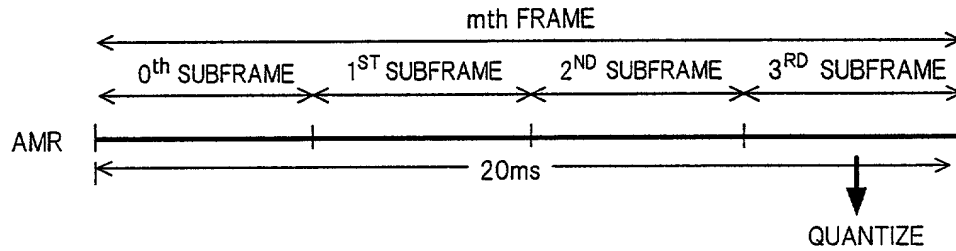


FIG. 4A**FIG. 4B****FIG. 6**

(a)	SUBFRAME NO. / FRAME NO.			
	G.729	0/ n	1/ n	0/ $(n+1)$
	AMR	0/ m	1/ m	2/ m
CORRESPONDS TO NUMBER OF BITS OF PITCH-LAG CODE IN EACH SUBFRAME				
(b)	PITCH-LAG CODE (NUMBER OF BITS)			
	G.729	8	5	8
	AMR	8	6	8
CORRESPONDS TO NUMBER OF BITS OF ALGEBRAIC CODE IN EACH SUBFRAME				
(c)	ALGEBRAIC CODE			
	G.729	17	17	17
	AMR	17	17	17

FIG. 5

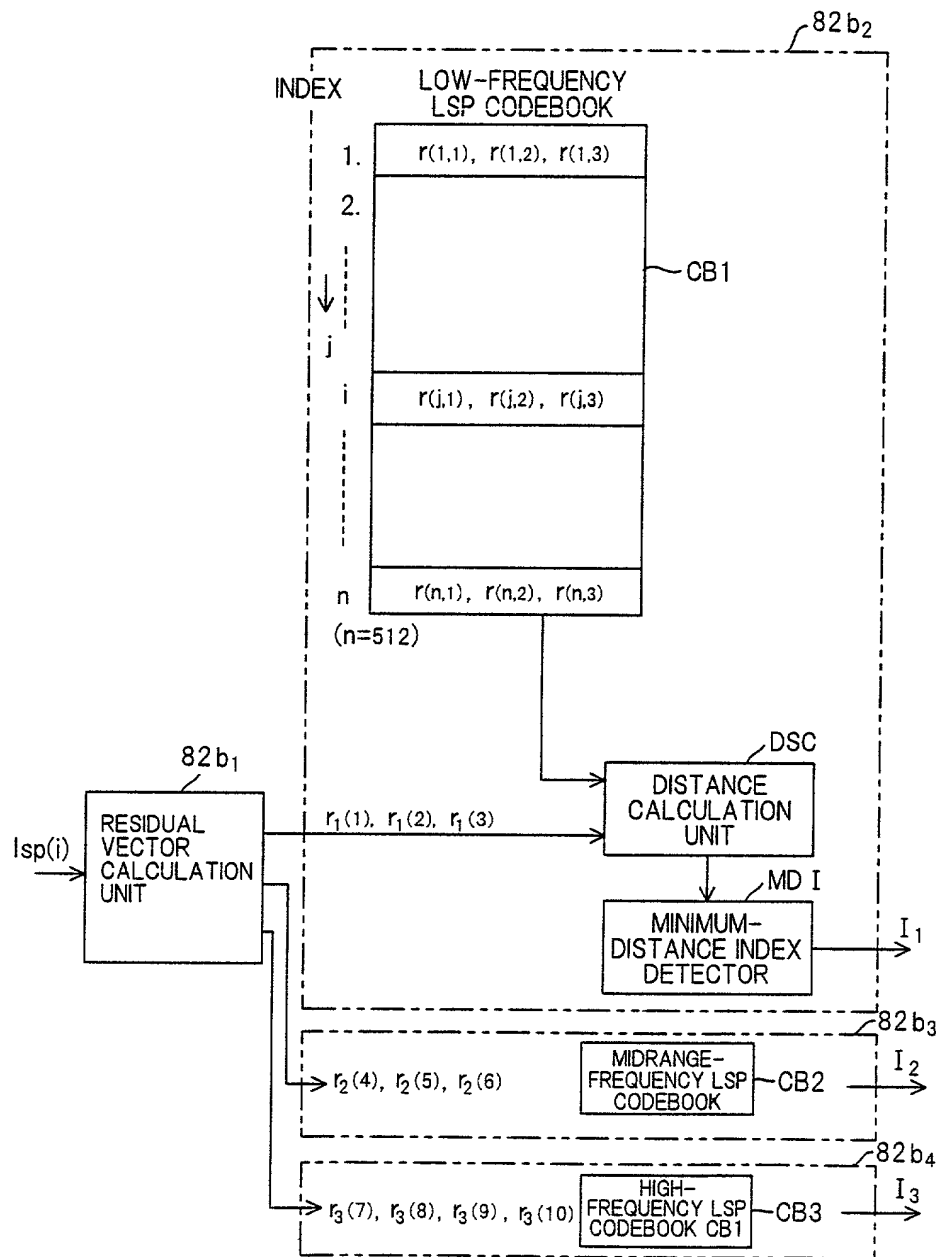


FIG. 7A

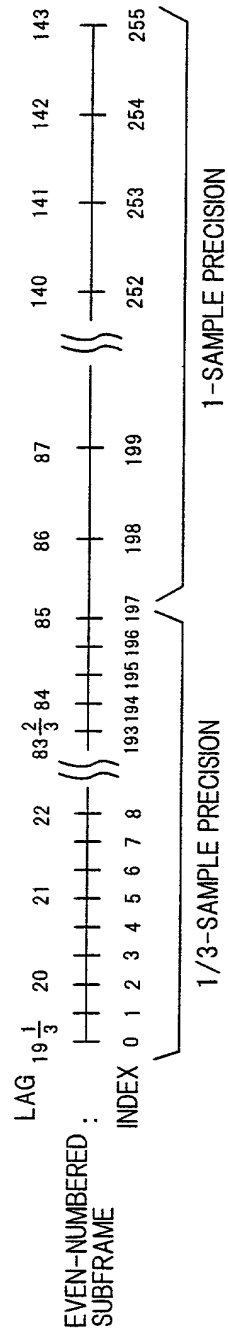


FIG. 7B

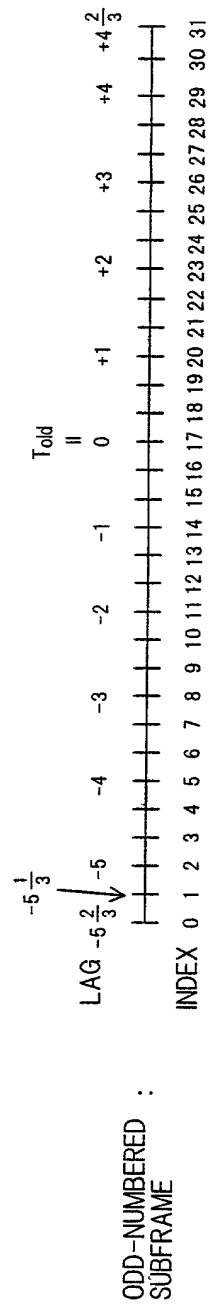


FIG. 8A

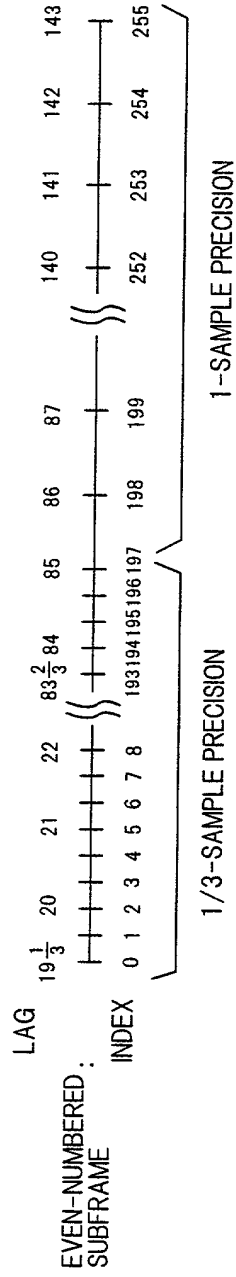


FIG. 8B

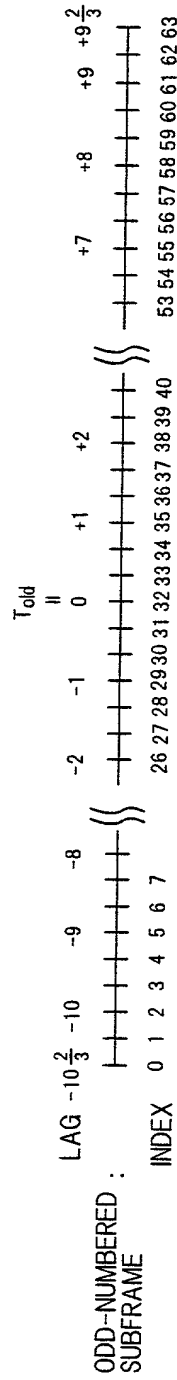


FIG. 9

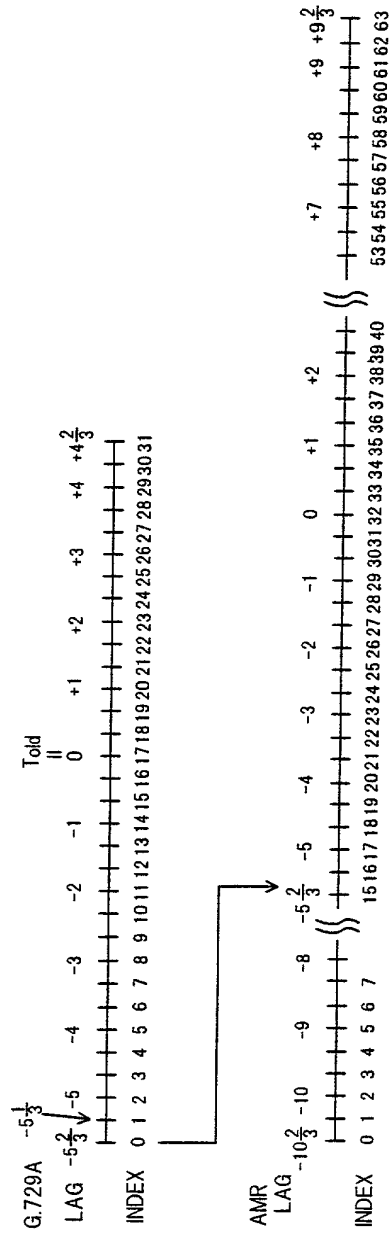
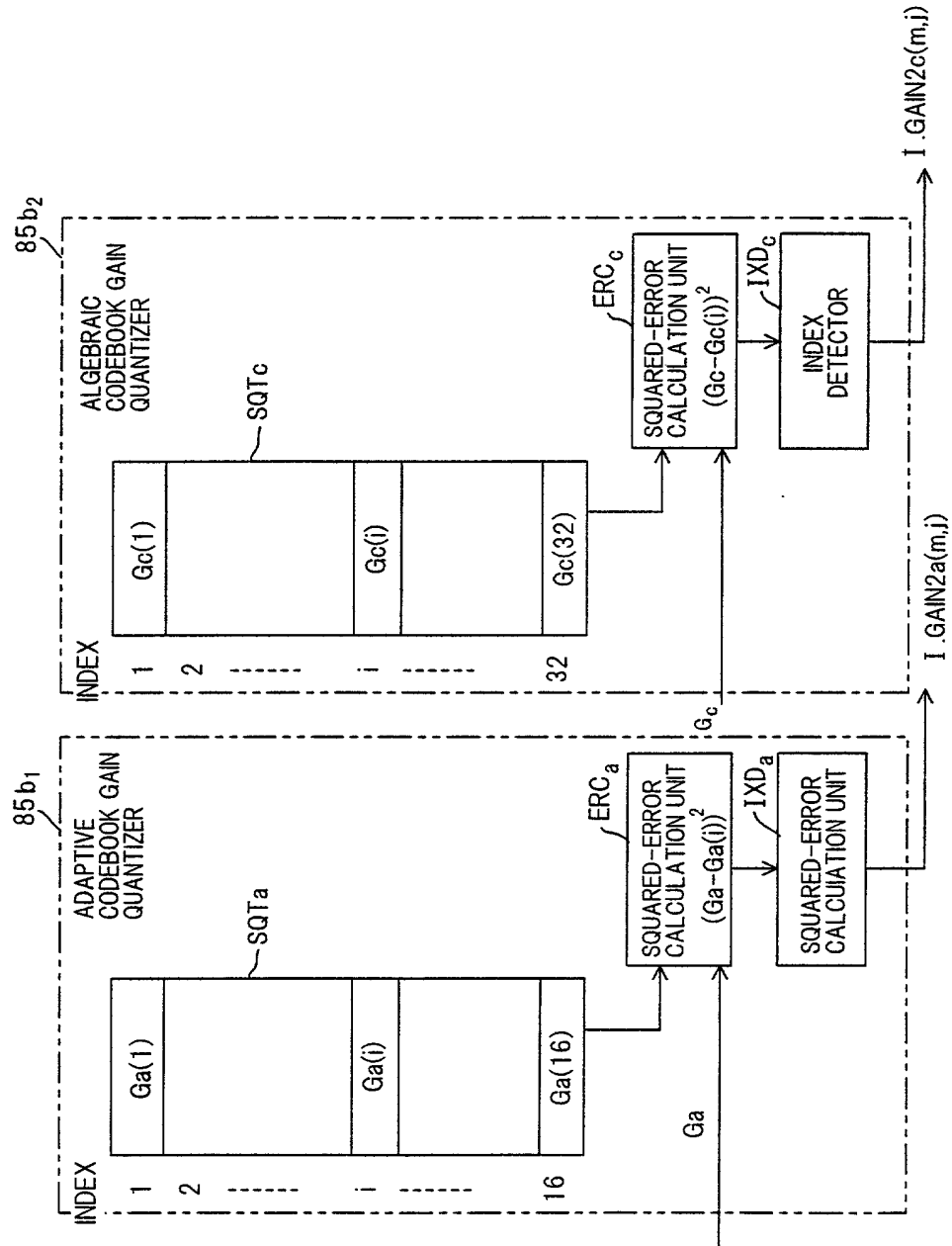
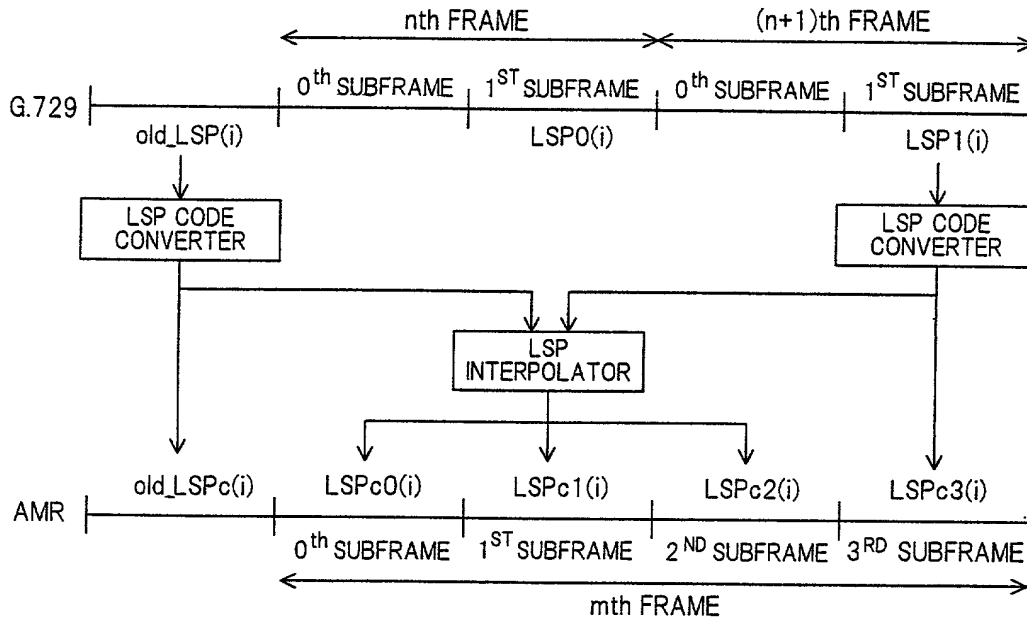


FIG. 10



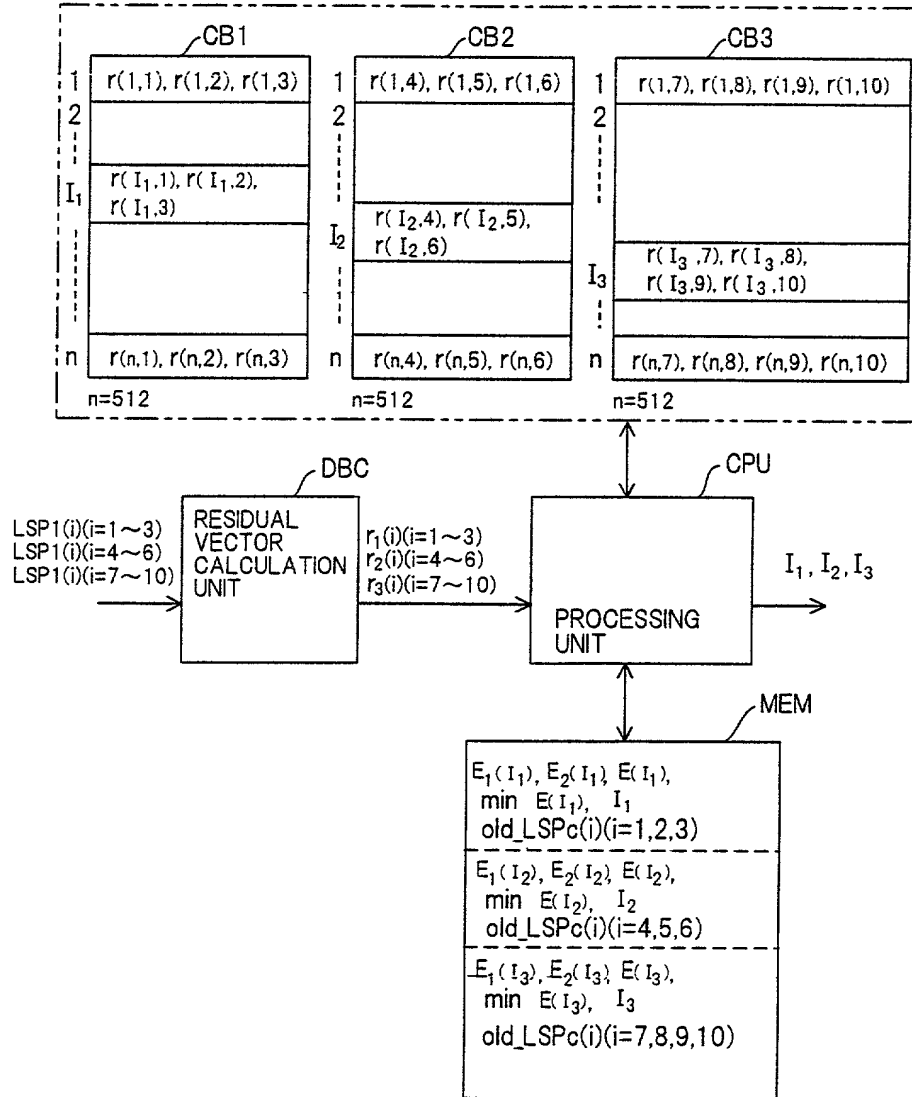
10/29

FIG. 11



11/29

FIG. 12



12/29

FIG. 13

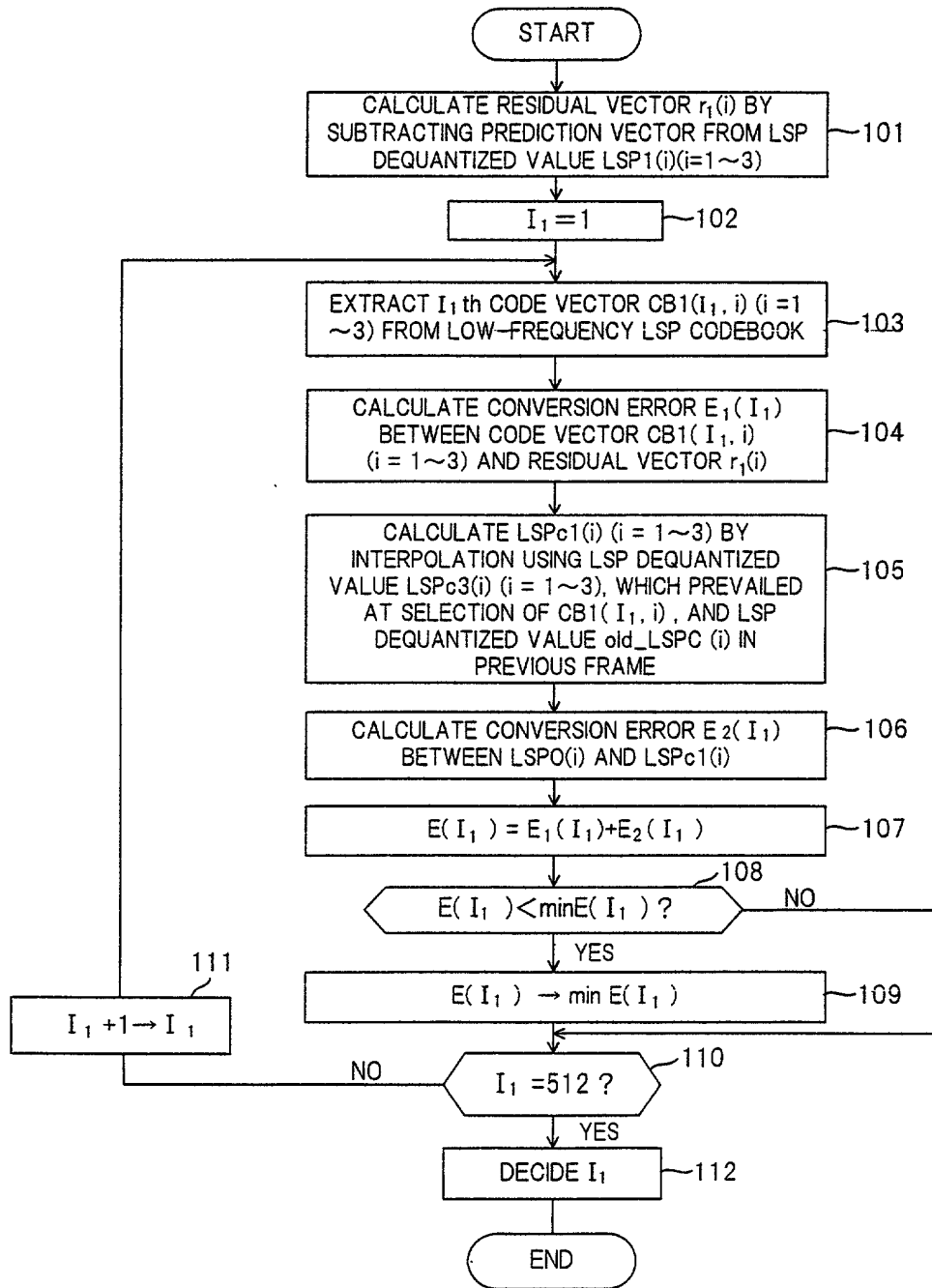


FIG. 14

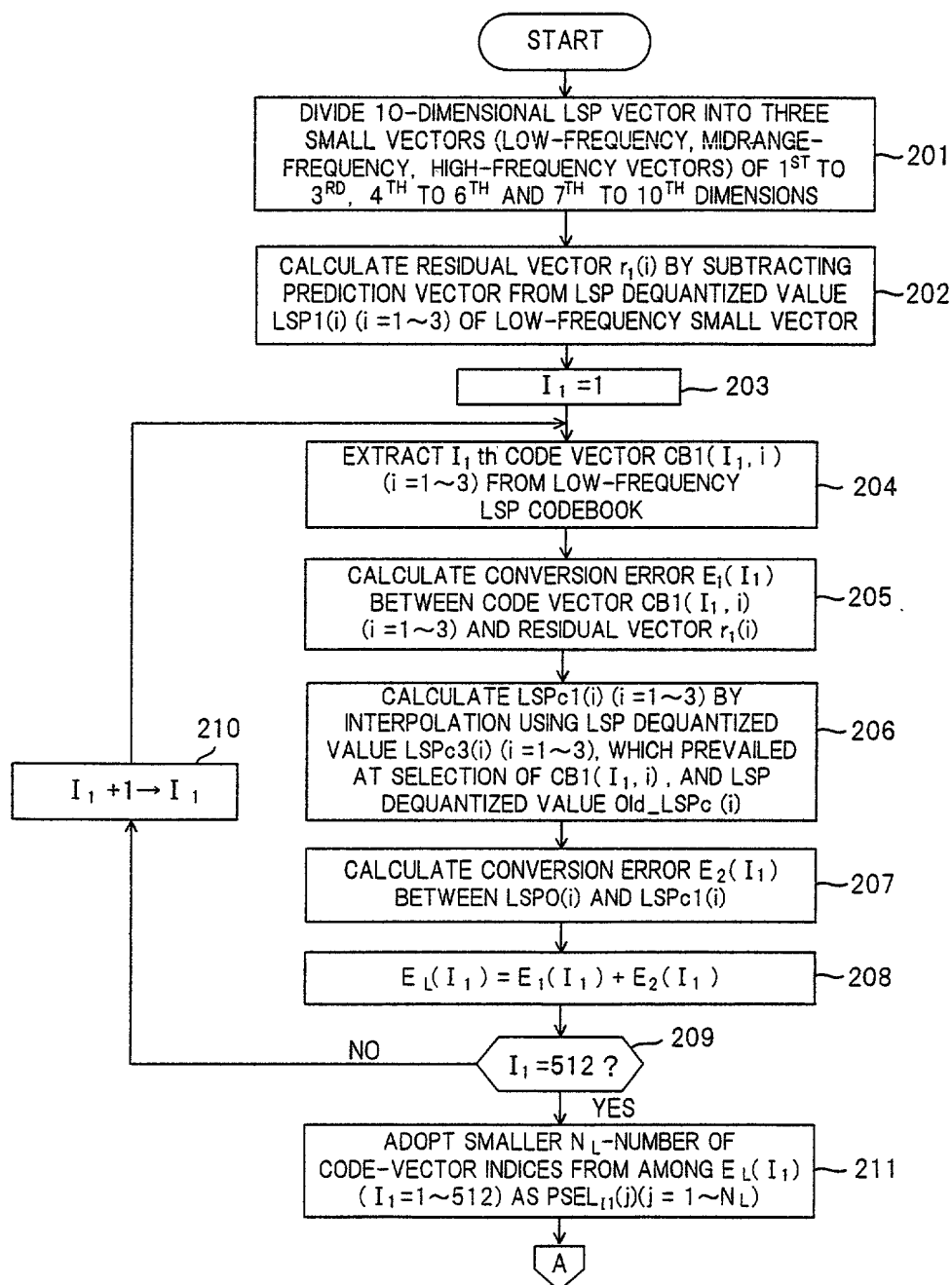


FIG. 15

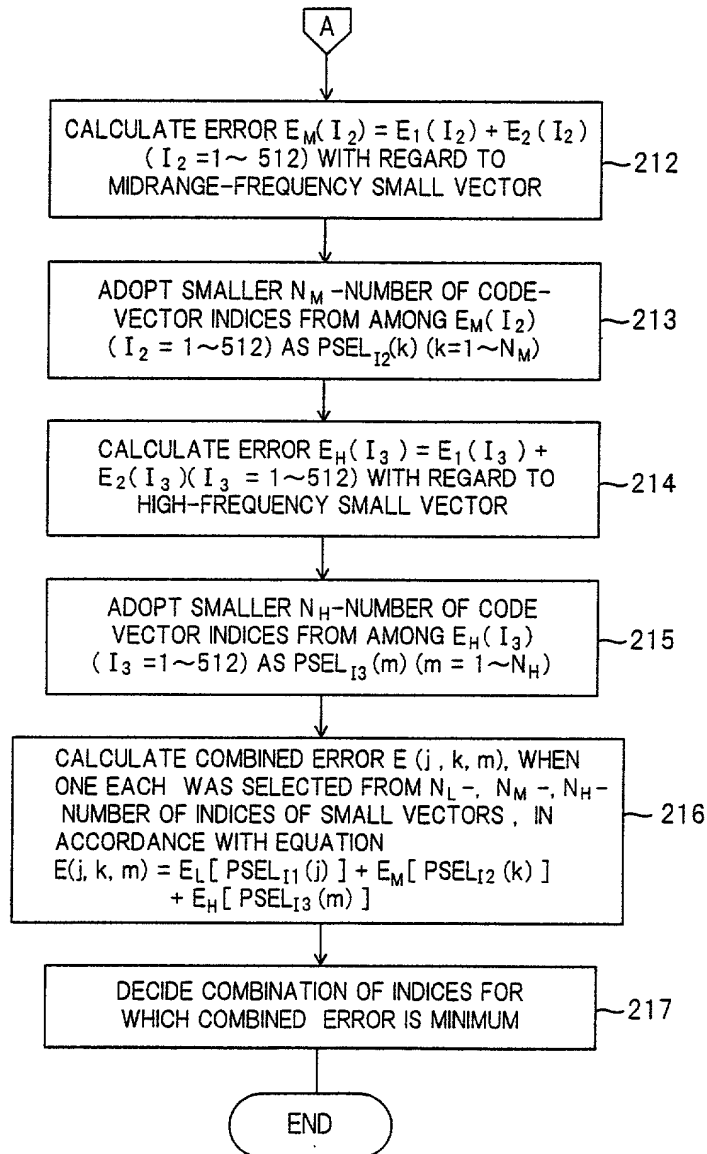


FIG. 16

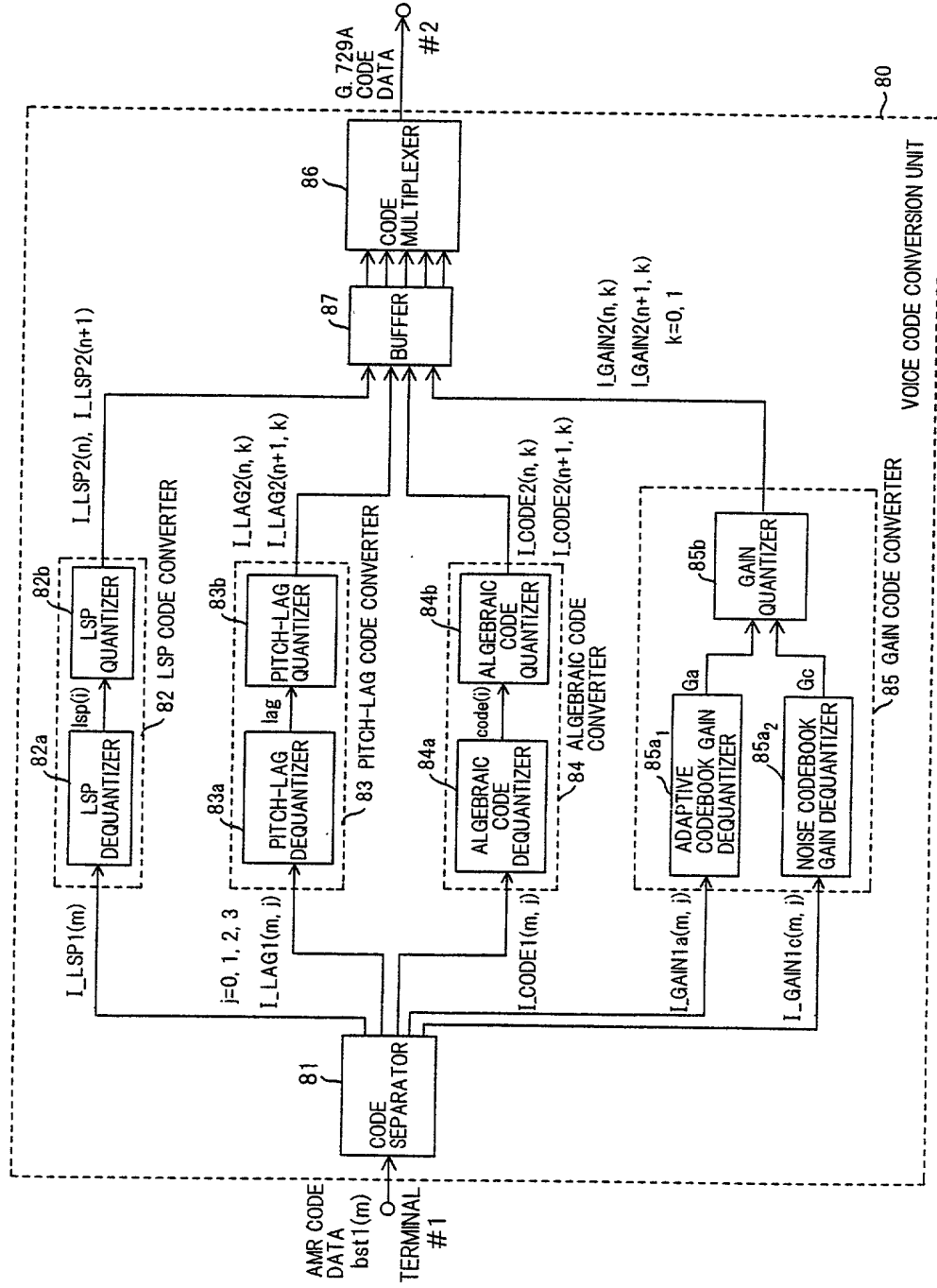


FIG. 17

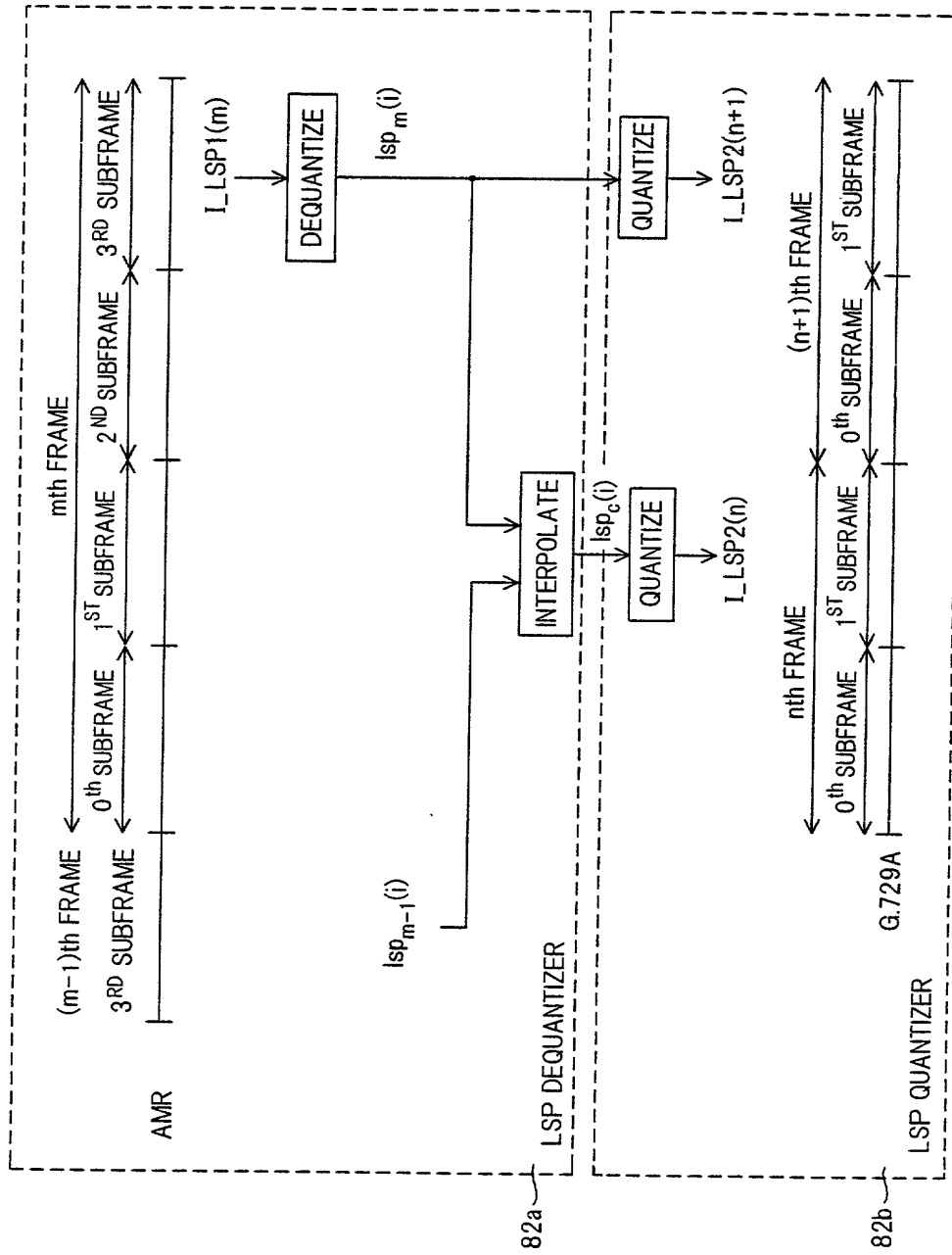


FIG. 18

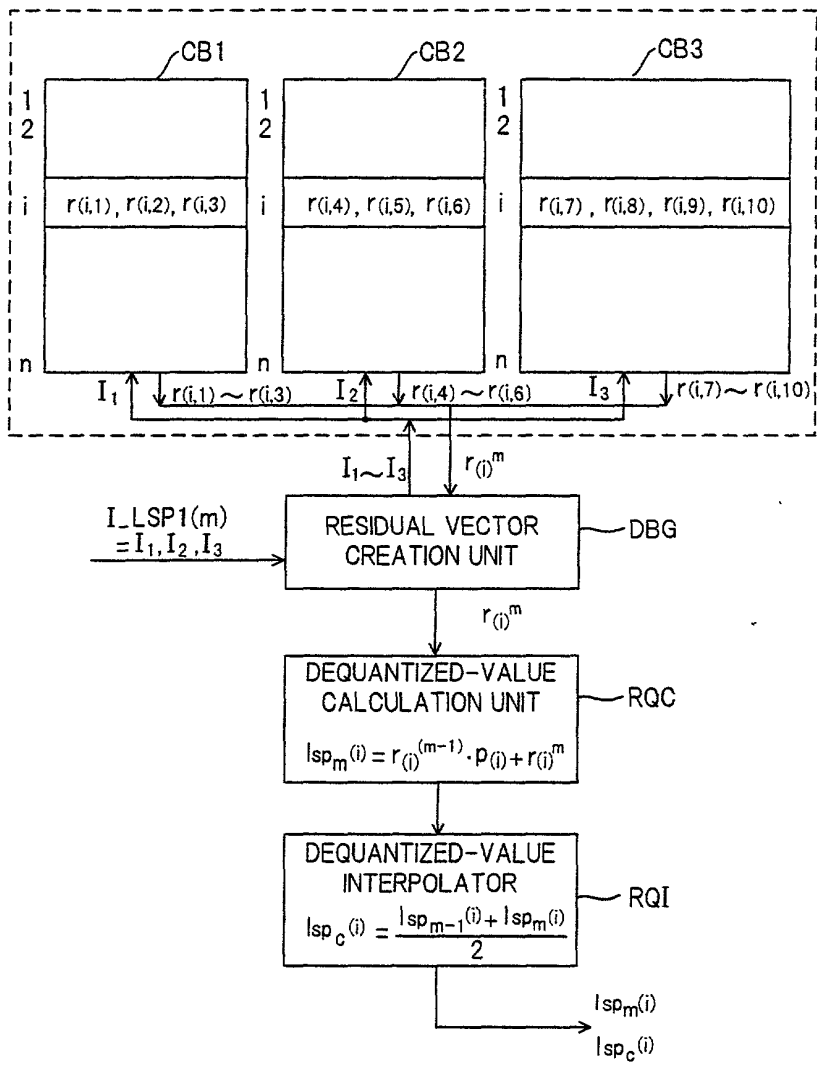


FIG. 19

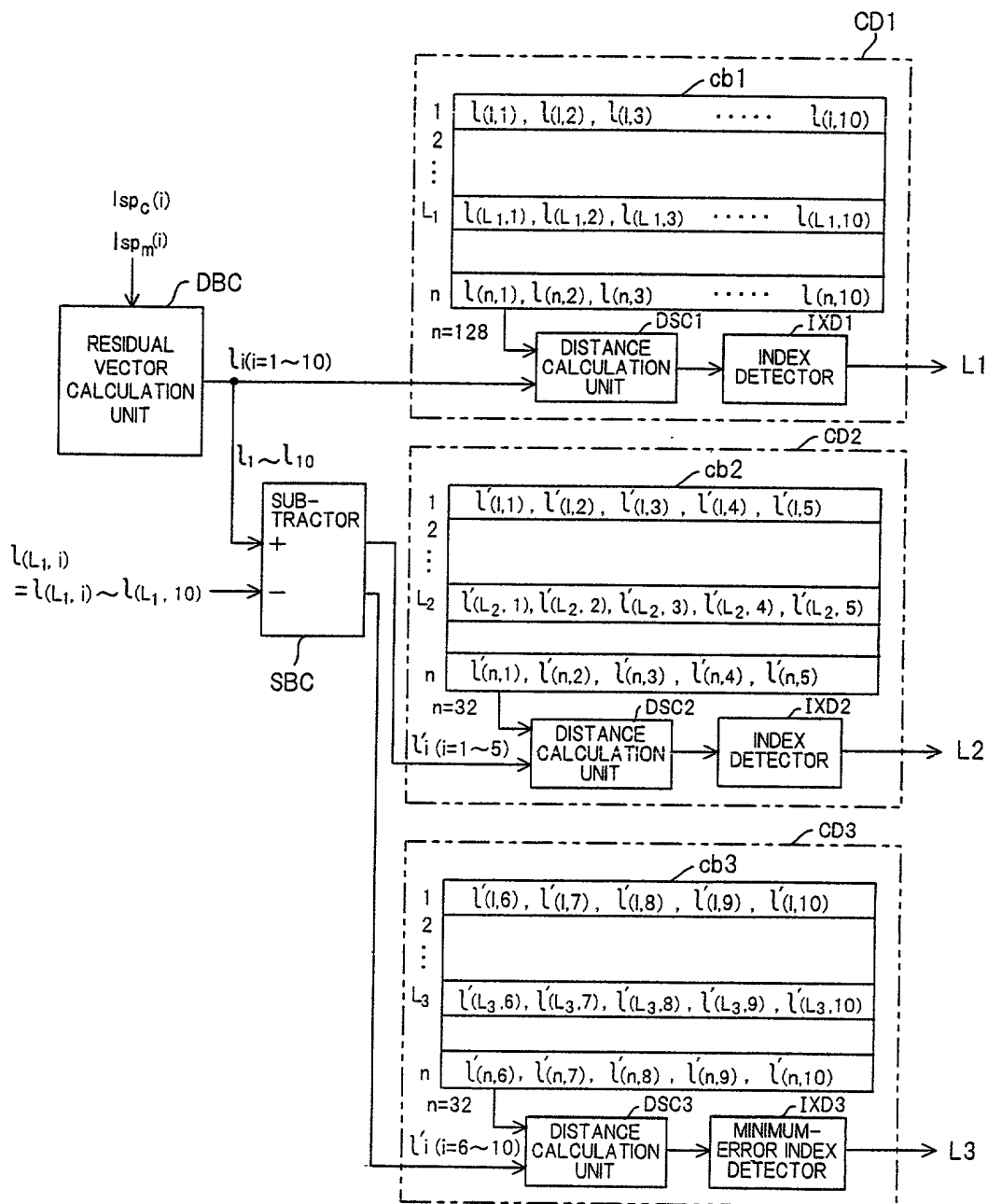


FIG. 20

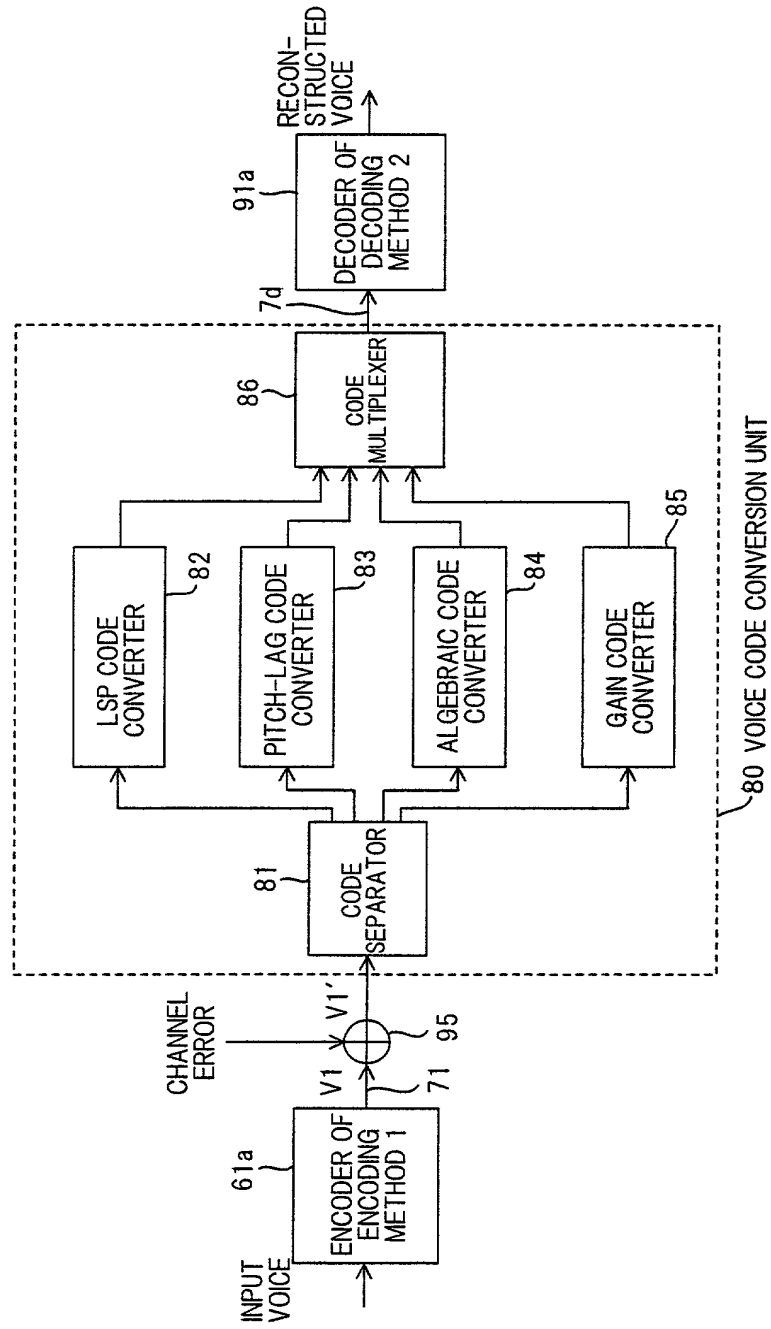


FIG. 21

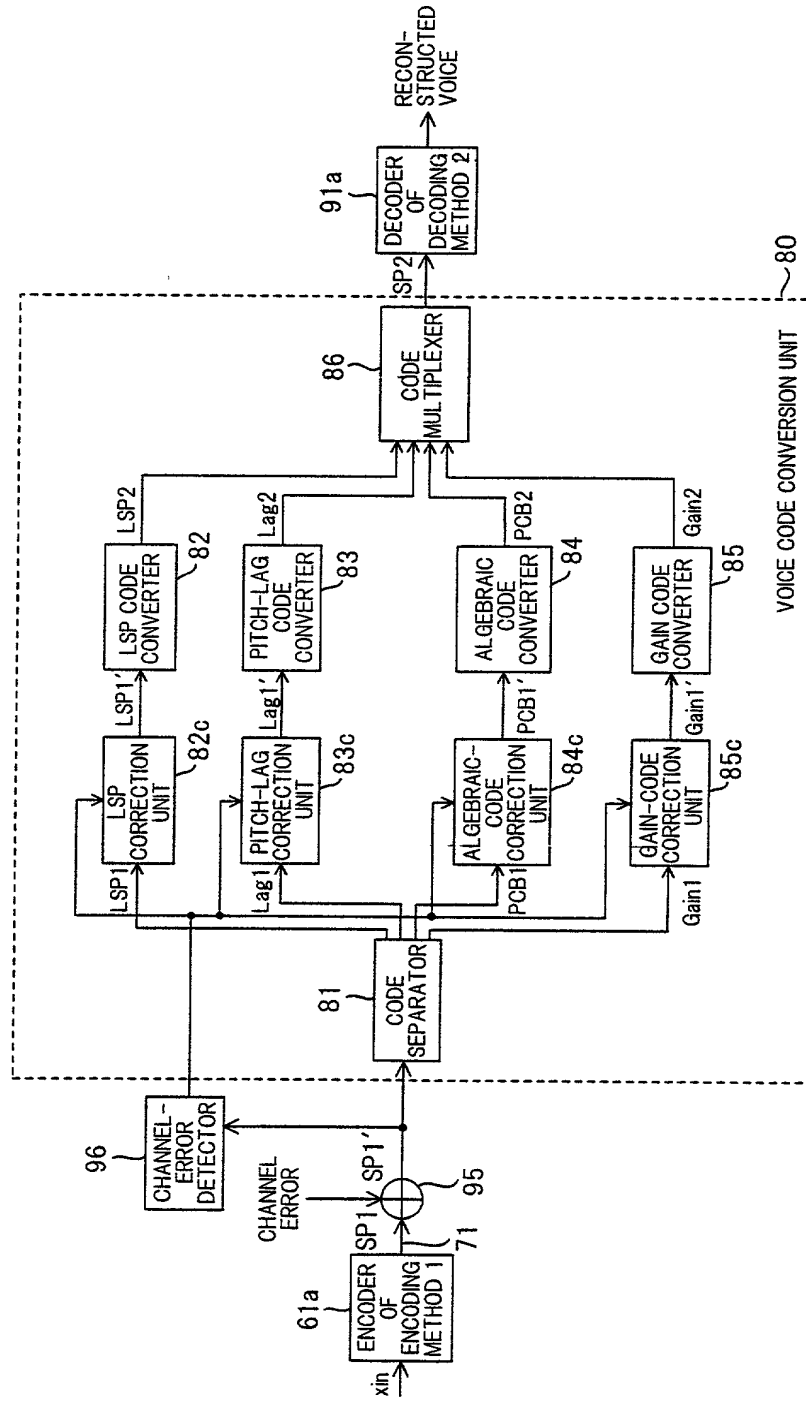
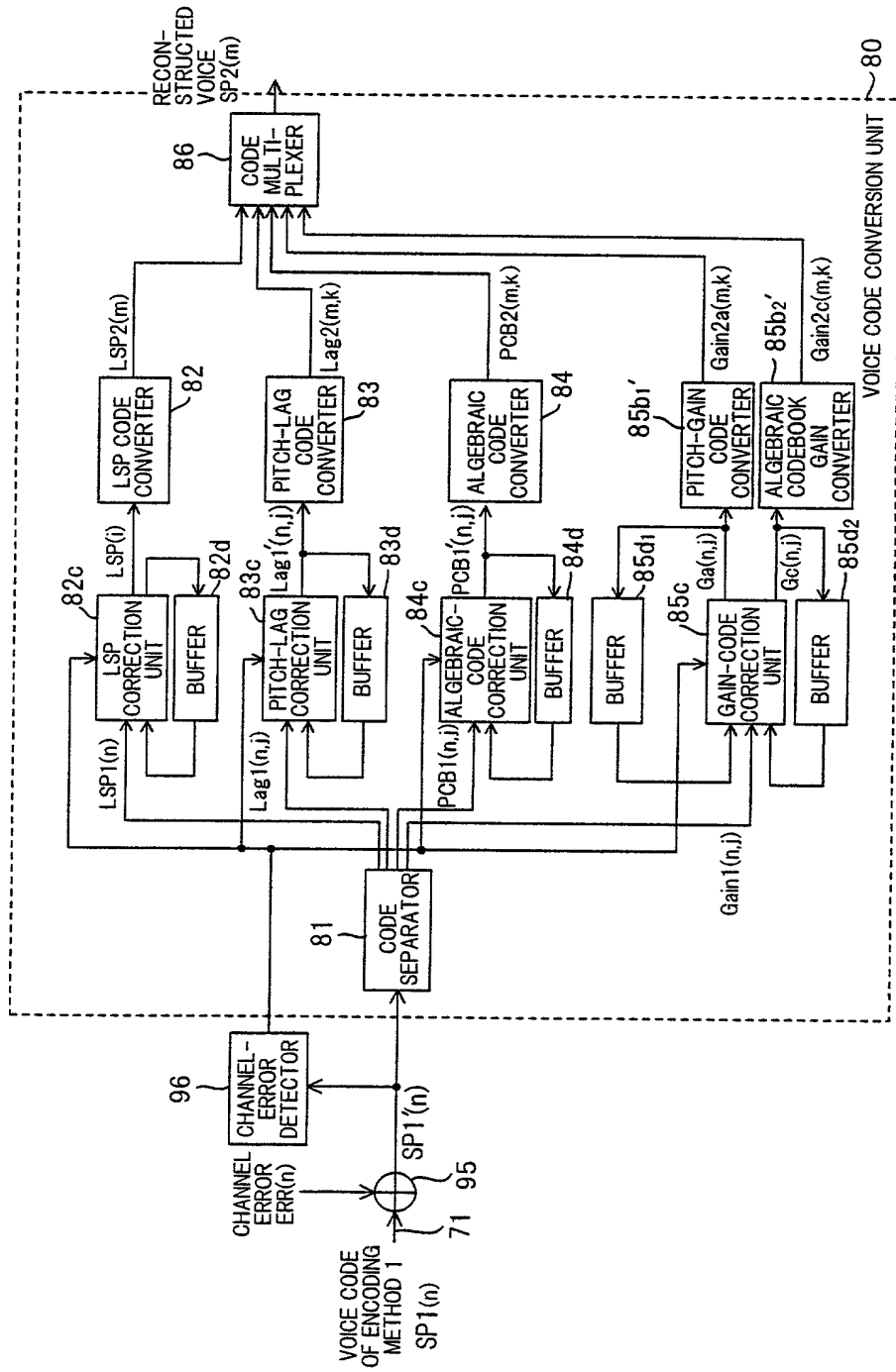


FIG. 22



The diagram illustrates a speech synthesis system architecture. It begins with an **INPUT SIGNAL X** entering a **LPC ANALYZER (1)**, which outputs **LPC COEFFICIENTS (2)**. These coefficients pass through a **PARAMETER CONVERTER (3)** to produce **LSP PARAMETERS (3)**. These parameters are then processed by an **LSP QUANTIZER (3)** to yield **LSP DEQUANTIZED VALUES (4)**. These values are interpolated by an **LSP INTERPOLATOR (4)** to produce **LSP INTERPOLATED VALUES (5)**. These interpolated values are then processed by a **PARAMETER REVERSE CONVERTER (5)** to produce **SYNTHESIS FILTER COEFFICIENTS (6)**. These coefficients are fed into an **LPC SYNTHESIS FILTER (6)**. Simultaneously, a **SOUND-SOURCE SIGNAL** is added to the output of the LPC synthesis filter at a summing junction (11) to produce the **RECONSTRUCTED VOICE (9)**. The reconstructed voice signal is fed into an **ERROR-POWER EVALUATION UNIT (10)**. The output of the error-power evaluation unit is fed back into a **GAIN QUANTIZER (12)**, which also receives **Ga** and **Gc** signals. The gain quantizer outputs a **GAIN CODE**. This gain code is fed into an **ADAPTIVE CODEBOOK (7)** and an **ALGEBRAIC CODEBOOK (8)**. The adaptive codebook also receives an **INDEX** signal (1, 2, ..., L-1, L) and outputs **ADAPTIVE CODEBOOK GAIN DEQUANTIZED VALUES (13)**. The algebraic codebook outputs **ALGEBRAIC CODEBOOK GAIN DEQUANTIZED VALUES (14)**. These two sets of values are combined at a summing junction (11) to produce the **SOUND-SOURCE SIGNAL**. The adaptive codebook also outputs **PITCH-LAG CODE** and **ALGEBRAIC CODE**. The pitch-lag code is fed into a **LINE ENCODER (15)**, which also receives **LINE DATA** and outputs a **LINE CODE**. The algebraic code is fed into a **GAIN QUANTIZER (12)**, which also receives **Ga** and **Gc** signals. The gain quantizer outputs a **GAIN CODE**. The gain code is fed into the adaptive codebook and the algebraic codebook. The adaptive codebook also receives an **INDEX** signal (1, 2, ..., L-1, L) and outputs **ADAPTIVE CODEBOOK GAIN DEQUANTIZED VALUES (13)**. The algebraic codebook outputs **ALGEBRAIC CODEBOOK GAIN DEQUANTIZED VALUES (14)**. These two sets of values are combined at a summing junction (11) to produce the **SOUND-SOURCE SIGNAL**. The reconstructed voice signal is fed into an **ERROR-POWER EVALUATION UNIT (10)**, which outputs a signal to the gain quantizer (12).

FIG. 24 PRIOR ART

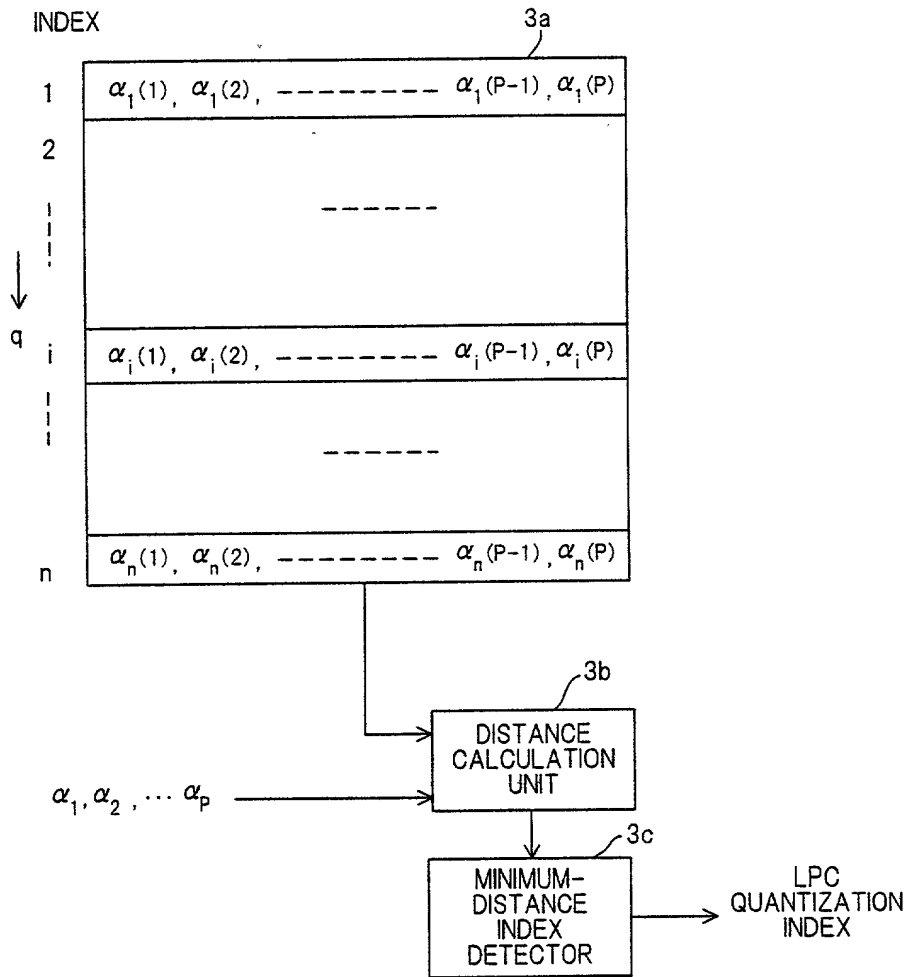


FIG. 25 PRIOR ART

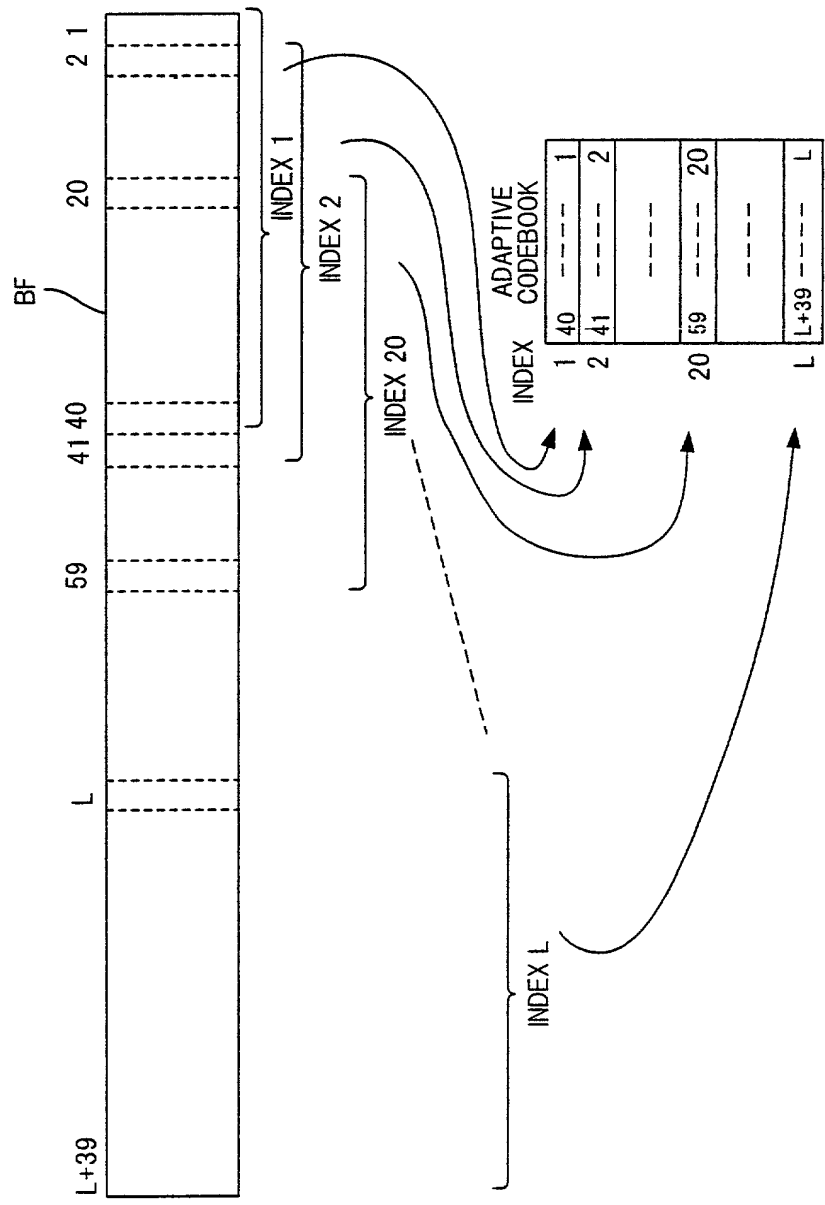


FIG. 26 PRIOR ART

PULSE SYSTEM	PULSE POSITION	POLARITY
1	0,5,10,15,20,25,30,35	+/-
2	1,6,11,16,21,26,31,36	+/-
3	2,7,12,17,22,27,32,37	+/-
4	3,8,13,18,23,28,33,38 4,9,14,19,24,29,34,39	+/-

FIG. 27 PRIOR ART

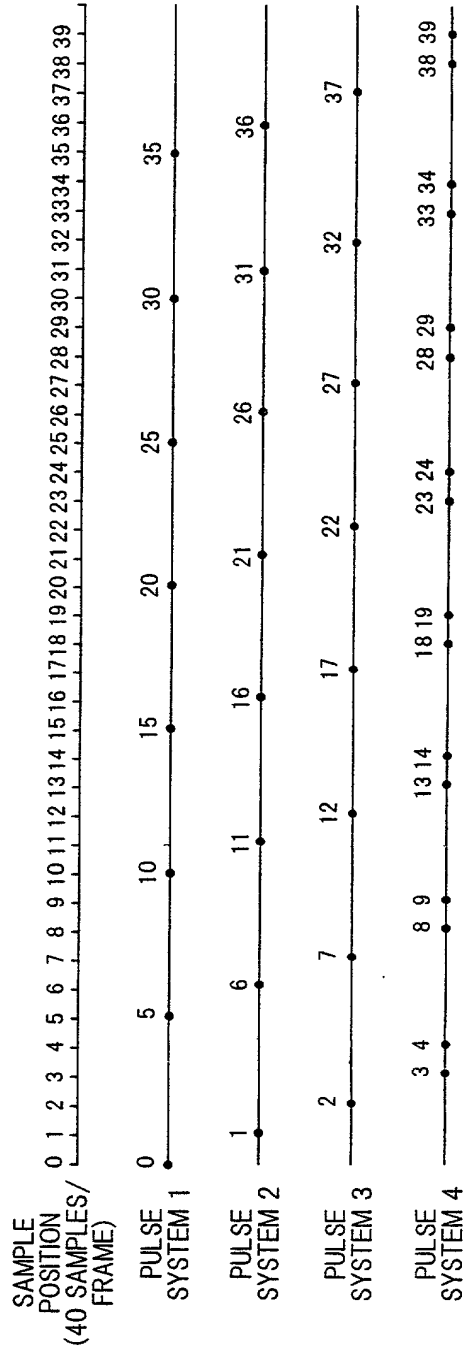


FIG. 28 PRIOR ART

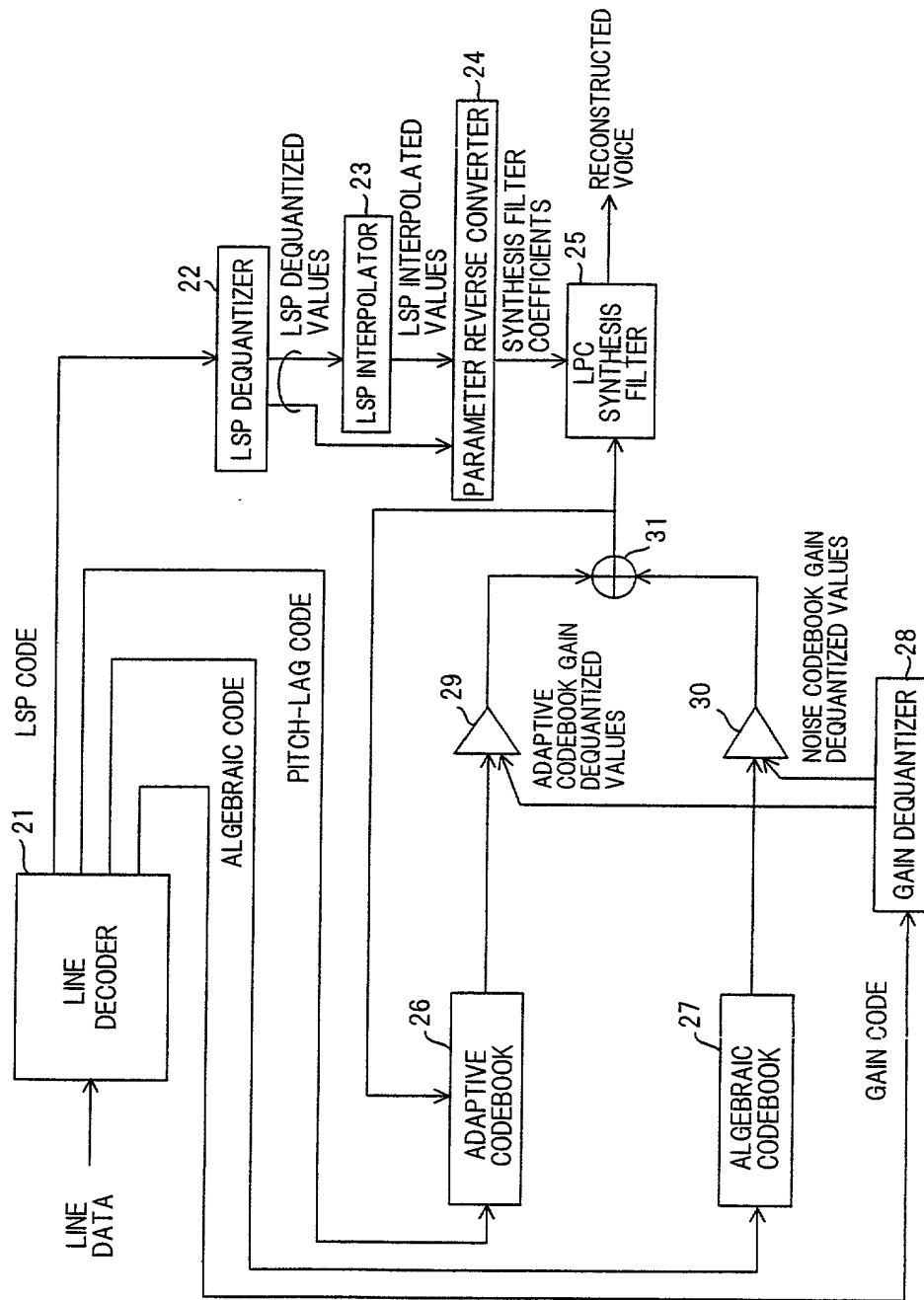


FIG. 29 PRIOR ART

	ITU-T G.729A	GSM-AMR
SAMPLING FREQUENCY	8kHz	8kHz
FRAME LENGTH	10ms	20ms
SUBFRAME LENGTH	5ms	5ms
NUMBER OF SUBFRAMES	2	4
BASIC DELAY	15ms	20ms
LINEAR PREDICTION DEGREE	10	10

FIG. 30 PRIOR ART

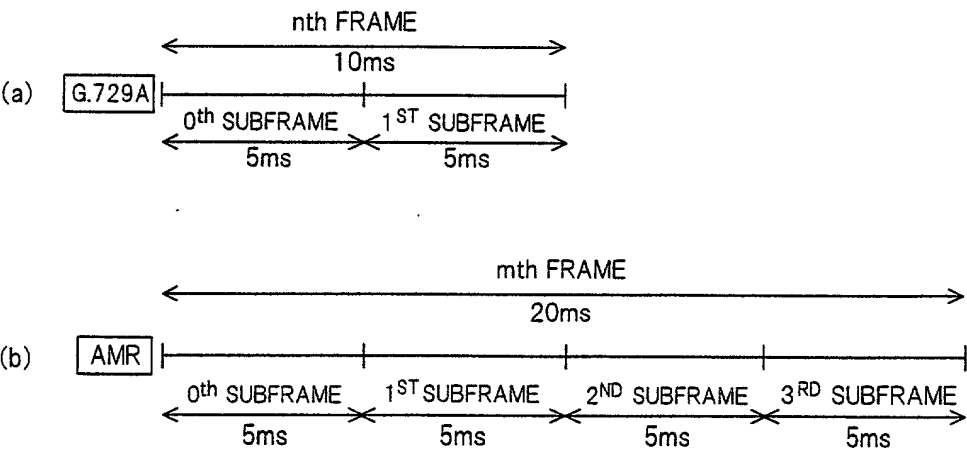


FIG. 31 PRIOR ART

	ITU-T G.729A	AMR (7.95-kbps MODE)
PARAMETER	BIT LENGTH (SUBFRAME/FRAME)	BIT LENGTH (SUBFRAME/FRAME)
LSP CODE	-/18	-/27
PITCH-LAG CODE	8+5/13	8+6+8+6/28
PITCH PARITY	1/1	—
ALGEBRAIC CODE	17+17/34	17+17+17+17/68
GAIN CODE	7+7/14	—
ADAPTIVE CODEBOOK GAIN CODE	—	4+4+4+4/16
ALGEBRAIC CODEBOOK GAIN CODE	—	5+5+5+5/20
TOTAL	80bit/10ms	159bit/20ms

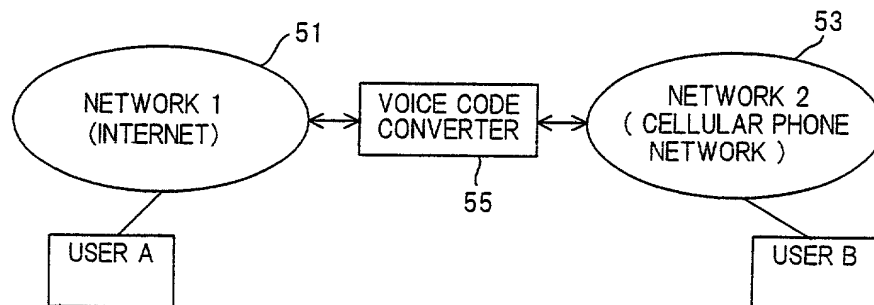
FIG. 32 PRIOR ART

FIG. 33 PRIOR ART

